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ABSTRACT

This paper discusses common causes of problems encountered with audio systems in distance learning networks and offers practical suggestions for correcting the problems. Problems and discussions are divided into nine categories: (1) acoustics, including reverberant classrooms leading to distorted or garbled voices, as well as one-dimensional audio output; (2) correct microphone usage, including creating a natural sound environment with carefully placed microphones; (3) correct loudspeaker usage, including "zoning" loudspeakers and balancing speaker output; (4) acoustic echo, including microphone pick-up of received audio off loudspeakers and re-transmission of audio back to the originating site; (5) outside noises, including blower/fan noises, buzz from fluorescent lighting, noises from outside the room, and paper shuffling; (6) sound system problems, including electronic and acoustic feedback; (7) telephone system feedback, including squealing and howling whenever a phone line is introduced and phone coupler problems; (8) other telephone system concerns, including listener fatigue from processing out line noise and compensating for thin, tinny audio; and (9) satellite echo, i.e., delayed repetition of voices over the satellite. Three diagrams are included to illustrate reverberation and its effects on transmitted audio, acoustic coupling generated by an open microphone and speaker, and the use of an acoustic echo canceller. (DLS)

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Improving Audio Quality in Distance Learning Applications

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Introduction

If you're like most educators, distance learning represents a fun and interesting challenge for you. The logistics of dealing with multiple locations and the problem of keeping everyone interested in your topic are your main concerns. And, if the distance learning network is designed correctly, they should be your *only* concerns.

The problem is that in a large number of distance learning networks, careful attention is paid to the video side—camera locations, position of the lighting, ability to view monitors, etc.—but little attention is paid to the audio, which carries the bulk of your message (unless you're conducting a final exam on sign language or lip reading!). Does your distance learning network have any of these problems?

- "Bottom of the barrel" sound (the rooms sound hollow and "boomy")
- Feedback/squealing in the PA system
- Feedback/squealing whenever you bring a phone line into the conference
- Voices are distorted or garbled (difficult to hear or understand participants)
- Fans or blowers overpower voices
- Some voices are too faint, others too loud
- Background noise is really annoying
- Acoustic echo (you hear your own voice coming back to you, or your students hear their own voices coming back to them)
- Satellite networks: your students hear their voices coming back to them over the satellite, with a long delay
- Telephone networks: the audio is thin, tinny, difficult to listen to for long periods of time

You do not have to put up with bad audio in your distance learning network!

This paper will discuss the most common causes of the problems encountered with audio systems, and practical suggestions for correcting the problems.

It's (Almost) All in the Acoustics

Here is a bold (but true) statement: *Most audio problems in your distance learning network are caused by poor acoustics in the classrooms.*

With only a few exceptions, the majority of your problems can be traced to the hard surfaces you'll find in the classroom—walls, floors, ceiling, windows, and tables. A room with lots of parallel hard surfaces is said to be highly *reverberant*, meaning a sound introduced into the room will bounce around for some time before decaying beyond your ability to hear it (see Figure 1).

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"But," you say, "we've had reverberant classrooms ever since schoolhouses were first designed. What makes them a problem in distance learning?"

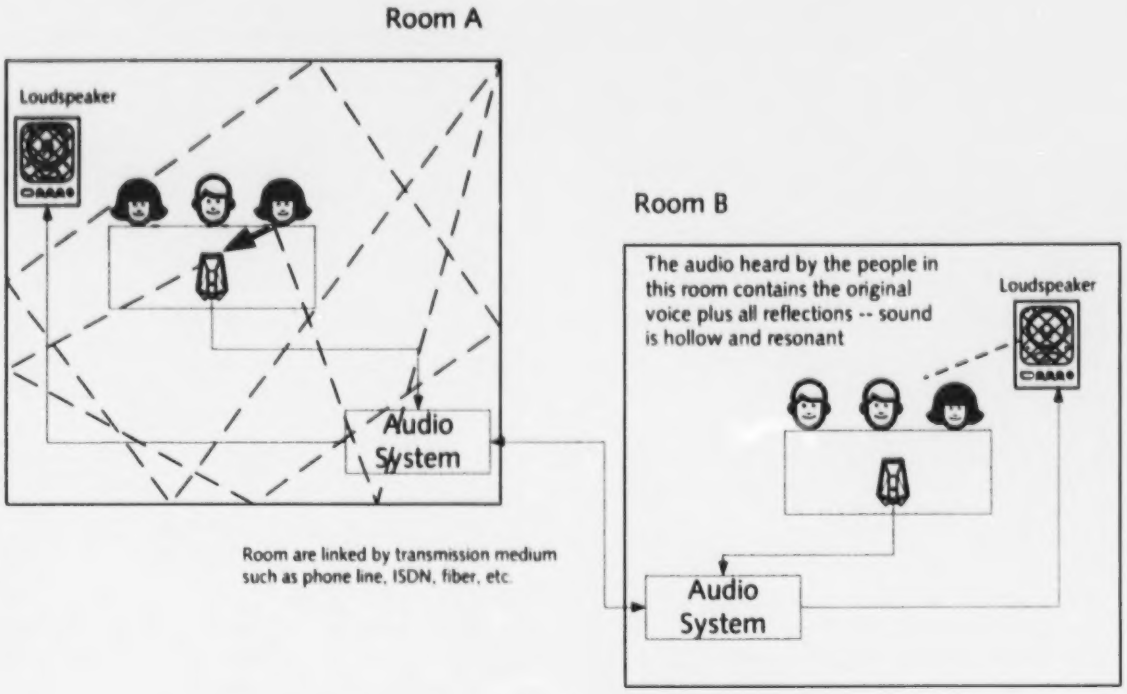
Room A
Loudspeaker
Audio System
Room B
The audio heard by the people in this room contains the original voice plus all reflections -- sound is hollow and resonant
Loudspeaker
Room are linked by transmission medium such as phone line, ISDN, fiber, etc.
Audio System

Figure 1: Reverberation and its effects on transmitted audio.

Reverberant classrooms only cause minor problems for the people in them (the biggest problem being fatigue) because the position of our ears, combined with the processing power of our brains, permit us to ignore the bouncing audio and concentrate on the source of the original sound (hopefully, the teacher). In other words, we hear three-dimensionally. A microphone, on the other hand, simply picks up everything it "hears" and sends it to the other classrooms. The original sound is given the same treatment as a reflection of the sound because the microphone cannot differentiate between the two. The students at the other sites do not have the benefit of 3-D hearing because ALL sounds are reaching them from a single location (the loudspeaker). The result is that the students in the distant locations hear everything—original audio plus reflections—at one level (the loudspeaker's volume) which gives them distorted, garbled sound.

There are a number of "fixes" for reverberant rooms. The best "fix" is acoustic treatment: acoustical ceiling tiles, carpeted floors, sound-absorbing panels on the walls, and draperies over the windows. If you don't have the budget for sound panels, carpets or draperies, try tacking a few blankets to the walls and tossing some throw rugs on the floor. You'll be surprised how much "quieter" the room gets. You should invest in draperies no matter what else you do: they will dampen sound, and your video will look better without interference from sunlight.

There are also a couple of electronic "fixes" that could help:

- An automatic (or "gated") microphone mixer reduces the number of times reflected audio can reach a microphone by turning idle mics off (the mixer must have a "threshold" setting that sets the "on" level above the level of the reflected sounds). Even if you have acoustic treatment in the room, you may want an automatic mixer to keep background noise at a minimum. Automatic mixers will be discussed in more detail below.

- Microphone processors can distinguish between the original voice level appearing at a microphone and lower-level reflections, and adapt to prevent the lower level audio from being passed through the system. Mic processors can also remove background noise from the audio. (However, you must have a processor for EACH microphone in the room, which can get expensive.)

Correct Microphone Usage (or Why Don't We Sound as Good as the People on TV?)

Television has given us an inaccurate impression of how "teleconference" audio should sound:

- In the TV world, people in another room (or even hundreds of light years away) sound great, and no microphones are seen anywhere.

- On the evening news or the nighttime talk show monologue, the announcer sounds terrific, and he or she is not holding a microphone.

The answer to the first example is easy: *it's fiction!* The second example simply reflects good camera angles concealing a boom microphone . . . or if you look closely at the announcer, he or she may be wearing a lapel (lavaliere) microphone. In television, great care is taken to conceal microphones because the sight of them would distract from the desired appearance of the sets and actors. However, microphones *are* present, and there are a lot of them in use.

Another important concept: sound is carefully controlled in TV studios. Walls are non-parallel, sound stages are acoustically treated, heavy drapes surround sets (both to minimize reverberation and cut down on background noise) and a number of microphones are placed in strategic locations for the best pick-up of sounds. Television studios also use highly directional microphones to pick up only the desired sounds while blocking out extraneous noise.

It probably isn't practical to convert your classroom to a television studio, but you can incorporate certain aspects of one, especially when it comes to microphone usage.

- Make sure you have an adequate number of microphones in the room. If you don't have enough microphones, it will be difficult to hear everyone (some people will sound faint and far-off while others are loud). If you have too many microphones, however, background noise can become overpowering and it's easier for the room to break into feedback. A good guideline is to have one microphone per 2 to 3 students (an automatic mixer can also help: see below).

- Get microphones off desks and tables, but not too far away from the people! Low-profile "boundary" microphones that sit on a tabletop are attractive and work well for a corporate conference room, but in the paper-laden classroom, these microphones are bound to get covered (or rustling papers will be louder than the students' voices). Podium (or "gooseneck") microphones are best: these microphones involve a small stand mounted to the desk, and a flexible "neck" with a small microphone positioned about a foot above the table. Another option is to suspend microphones from the ceiling, with the mics "pointed" at the students. If you do this, make sure students don't have to look up in order to talk into the microphone; position the microphones a couple feet above and *in front* of the desks. Microphones that are flush-mounted on the ceiling *can* work, but also create problems by being too far away from the students (when a microphone is more than a couple feet away from the mouth, the "bottom of the barrel" effect kicks in). Another problem with flush-mounted ceiling microphones is the possibility of 'mechanical coupling,' where the mics pick up vibrations from the air handling system, loudspeakers, or people walking on the floor above you.

- If your classroom has more than four microphones (and it should unless it's a small class), it's recommended to use an automatic microphone mixer. When a large number of microphones are on at once, especially in a reverberant room such as a classroom, the room's "gain" into the audio system can become too high. This excessive gain results in squealing or howling (feedback). Also, as mentioned earlier, an excessive number of "on" mics will increase background noise in the audio system. The automatic mixer will ensure that a minimum number of microphones are "on" at once. Experts in the field recommend an automatic mixer if four or more microphones are used in a room.

  Look for an automatic microphone mixer that will:

  - be able to handle the number of microphones you will be using (most mixers are "expandable" or can be cascaded).

  - not require a specific type of microphone—this can prove costly.

  - provide an "automatic gating threshold"—this means that a microphone will only turn "on" when the sound level exceeds a pre-determined level. The smarter mixers can automatically set the gating threshold above the background noise.

  - permit a "chair override" on one of the channels. This permits the instructor to take control of the mixer by simply speaking into his or her microphone.

  - allow one microphone to be "always on." If all mics turn off, that room's audio will go away (including background noise), making it sound as though the connection were cut off. Leaving one microphone on will provide a more natural sound (and is essential for the proper operation of acoustic echo cancellers, which will be discussed below).

Correct Loudspeaker Usage

A common mistake in distance learning or teleconferencing classrooms is the tendency to use only one or two loudspeakers to carry the audio from the other class sites. This can result in the following problems:

- When too few loudspeakers are used, it's necessary to crank up the volume, which places the room closer to a "feedback point."
- The students near the loudspeakers are in pain from the volume, while students who are far away from them can barely hear what's going on.

Using an adequate number of loudspeakers will allow you to keep volume at a reasonable level throughout, and will minimize the amount of loudspeaker audio that is picked up by microphones. The best way to distribute loudspeakers is to place them in the ceiling (or mount them along the walls) and control them with a separate power amplifier. The better amplification systems permit "zoning" of the loudspeakers, which is useful in lecture halls for amplifying the instructor's voice along with the audio from other sites. Keep the loudspeakers directly above the instructor at a lower volume to prevent feedback via the instructor's microphone.

One final note on loudspeaker placement: try to isolate loudspeakers from microphones as much as possible. Remember that a microphone will pick up all sounds in its vicinity, and audio coming out of a speaker will be treated as just another voice in the room.

Acoustic Echo (or Why Am I Hearing Myself? This Is Really Annoying!)

Anytime two or more sites talk to each other with "open" microphones and loudspeakers, the potential exists for acoustic echo. Acoustic echo is caused whenever a site's microphones pick up received audio off loudspeakers and re-transmit the audio back to the originating site (see Figure 2).

The pick-up path of the audio can be direct (straight line from loudspeaker to microphones) or indirect (bouncing around the room, then hitting the mics). Depending on the size of the room, or how reverberant the room is (reverberant rooms bounce the audio around for quite a while), a certain amount of delay will occur between the time the audio appears on the loudspeakers and when it reaches the microphones. This "room delay," combined with any delays introduced by the transmission path, will result in a delayed echo that could rival deep canyons or baseball stadiums. Unless you're a professional sports announcer, you can become extremely unnerved by this delayed repetition of your voice.

Fortunately, it's easy to fix acoustic echo! All it takes is a product called an acoustic echo canceller (AEC). The AEC is a bi-directional device which is placed into a room's sound system between the microphone mixer and transmission system, and the receive port and loudspeaker system (see Figure 3).

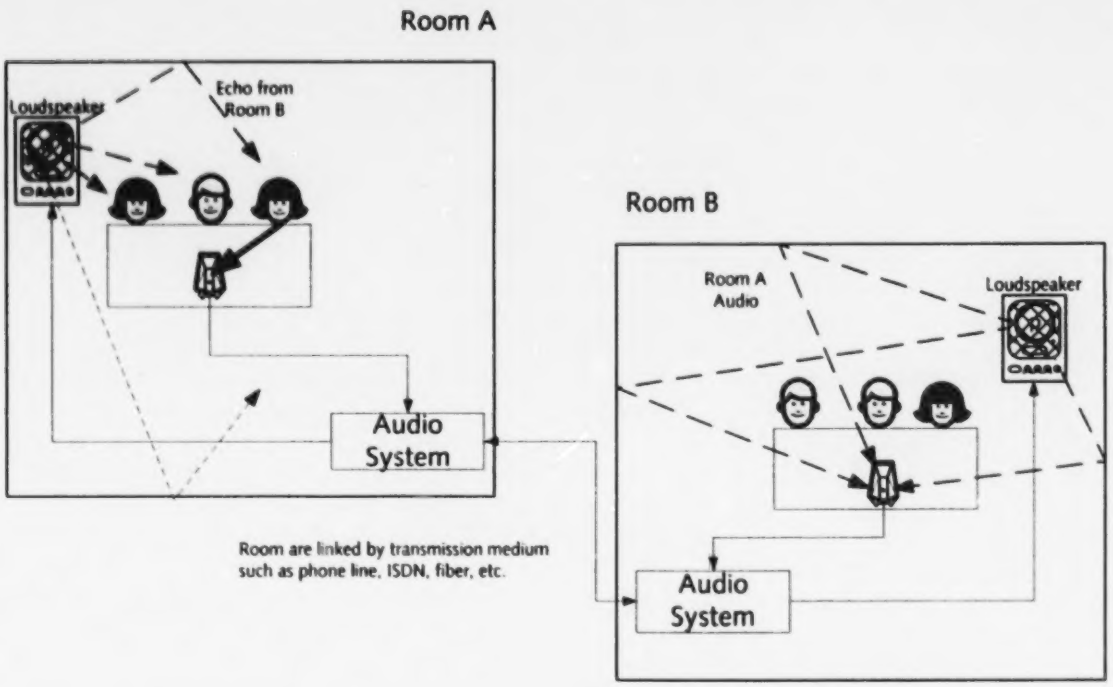

Figure 2: Acoustic coupling generated by an open microphone and speaker.

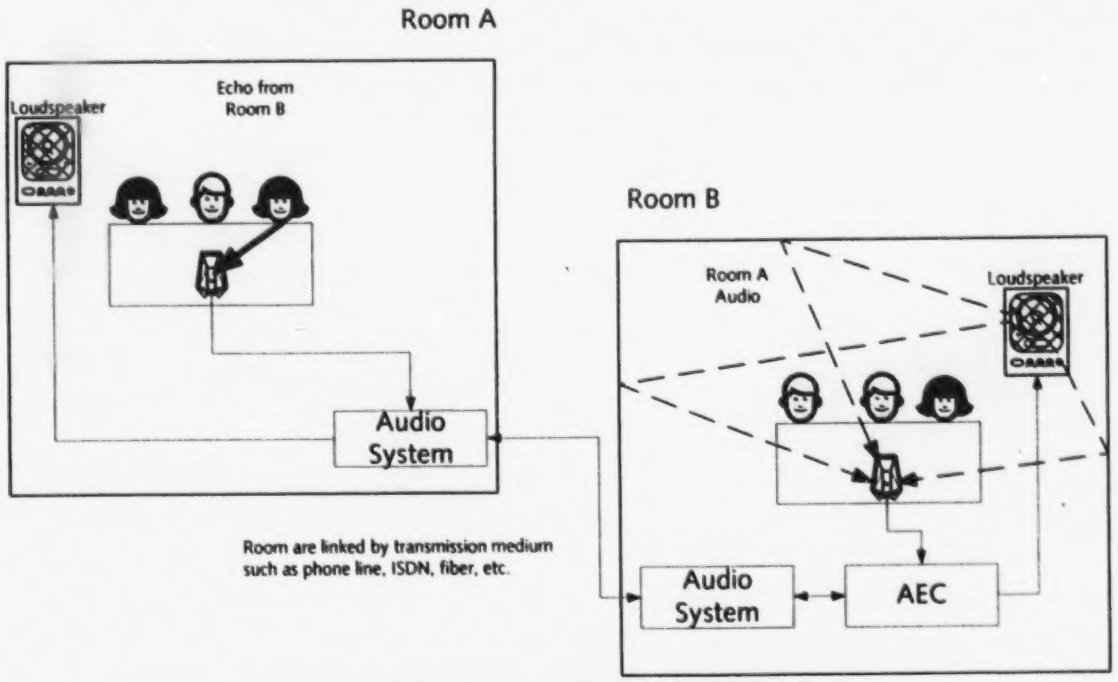

Figure 3: The AEC solution in Room B removes echoes from being transmitted back to Room A.

Acoustic echo cancellers use digital signal processing (DSP) technology to compare received audio with the audio being sent back down the transmission system. Any audio that has the same characteristics as the received audio is removed from the transmitted signal. Recent advancements in AEC technology have made these devices very effective in removing both "real-time" and delayed echoes.

An important note: an AEC in your room will do nothing to keep you from hearing the return of your own voice, because an AEC benefits the OTHER site(s) by removing their audio from the audio being sent to them. In order to eliminate acoustic echo at all sites, you will need to place an AEC at each location in the distance learning network.

Outside Problems (Noise That Isn't Part of the Sound System)

One of the most distracting things in a distance learning network is the extraneous noise that is picked up by microphones and sent to the other sites. This noise can take several forms:

- Blowers and other air handling systems
- Fans on computers, overhead projectors, etc.
- Buzz from fluorescent lighting
- Noises coming into the room from outside (talking, vacuum cleaners, traffic noise, etc.)
- Papers shuffling/rattling in front of microphones

Some of these problems aren't fixed easily. You may need to get the building contractor to install quieter air handling systems or have an electrical contractor change the lighting system. Computer/projector fan sounds and paper noises can be alleviated through correct microphone placement (see above). As for noises coming in from the hallway, one suggestion is to buy an "ON AIR—QUIET PLEASE" light and install it next to the classroom door in the hallway. (If your integrator does not have one of these lights, it can be purchased through a broadcast equipment dealer.) Traffic noises can be dampened somewhat with heavy curtains and other acoustic treatment (see above).

Sound System Problems (Squealing/Howling)

When a sound system starts squealing or howling, it is said to be in a feedback state. Feedback is caused by either an electronic or acoustic signal "feeding back" to its source, becoming amplified, and going through the cycle again.

- Electronic feedback is caused when an "output" signal is erroneously routed back to the sound system's input. This can also be caused by improperly adjusted telephone equipment (see below).

- Acoustic feedback is caused by having microphones too close to the loudspeakers that carry the microphone audio (local sound reinforcement).

Telephone System Feedback Problems

If your sound system starts squealing or howling whenever a phone line is introduced, the problem is not in the sound system but in the *interface* to the phone line. Most often, the wrong device has been used for bringing the phone call into the audio system. Telephone

"couplers," devices normally costing $300 or less, simply will not work for your application because they cannot adequately isolate the two sides of the telephone call. This inadequate isolation results in 'bleedthrough' of audio from the "send" side of the coupler to the "receive" side, and when this audio is amplified through your sound system, electronic feedback results.

Luckily, phone coupler problems are also easy to resolve. Throw out the couplers and replace them with "digital telephone hybrids," available through your systems integrator or a broadcast equipment dealer. These devices are the same products used at radio and TV stations to bring callers into talk shows. They are considerably more expensive than the couplers (they cost about $800 on the low end and $2500 on the high end) but won't introduce feedback into your audio system.

There is only one "trick" to using telephone hybrids: if the audio sent down the telephone line contains any of the callers' audio, feedback will result. The "trick" is to use what is called a "mix-minus" feed to the caller, which is a mix of all of the audio in your system *minus* the caller's audio. If your mixing system does not have mix-minus capability (most don't), you can either use a separate mixer for the phone line or buy a digital telephone hybrid with "automatic mix-minus" capability (these cost around $1500).

Other Telephone System Concerns

If your distance learning network is an audio-only system based on standard phone lines (rather than digital or satellite transmission), a significant problem can detract from the efficiency of your class: listener fatigue. Simply put, when you listen to a phone call for a long time, your brain has to work overtime just to process out the line noise and compensate for the thin, "tinny" sound of the line.

The best cure for phone line quality is to upgrade your distance learning network to another transmission system such as ISDN, ATM, or fiber optics. These systems offer higher "bandwidth," meaning voices sound natural, and transmission noise is eliminated because digital technology is employed. If your budget does not permit an upgrade of this type, you should at least try to maximize your system through the use of digital telephone hybrids combined with proper microphone and loudspeaker placement at each site.

Satellite Echo

Satellite echo is not a problem for the originating classroom, but can present a significant problem for the remote classes. The problem is this: the distant classrooms, receiving the class via satellite, "call in" their questions to the instructor. In order for the other classrooms to hear the questions, the students' audio is uplinked along with the instructor's voice. Normal satellite delays (usually around ¼ to ½ second) result in the students hearing a delayed repetition of their voices over the satellite. *Satellite echo has nothing to do with acoustic echo and cannot be fixed with normal AECs.*

At present, there is no "easy" fix to satellite echo. The most common methods of overcoming this problem are:

- Push to talk systems—these systems mute the incoming audio until the student has finished speaking; a "hold" is placed on the mute for about a second after the student

stops speaking so that the "tail" of the echo does not reach the classroom. The problem is that the "tail" mute often cuts off the first few words of the instructor's response as well. It's important to time your responses when dealing with push-to-talk systems. The delay in responding may feel unnatural to you but will aid in comprehension at the distant sites.

- Alternate receive audio systems—when the class goes to "Q&A" mode, the classes that are calling in to the uplink site simply turn off the audio from their satellite receiver and listen via the phone line instead. In order for this system to work well, each classroom must have a full "audioconferencing" set-up with digital telephone hybrids, microphones and loudspeakers. The uplink site must also be able to feed "mix-minus" audio to each of the sites calling in to the class.

It's important to note that the above "satellite echo" problem occurs only with satellite based networks using one-way video and two-way audio. If your satellite system provides two-way video and audio, it is easier to deal with satellite echo problems at the individual sites (a mix-minus system is employed at the uplink site). However, you must have very good acoustical treatment and AECs at each site, or acoustic echo generated at the distant classrooms will introduce a new form of satellite echo into the system—repeat delayed echoes of everyone!

Summary

Yes, it is hard to achieve good sounding audio in distance learning networks. There are many obstacles beginning with the room acoustics, continuing with the electronics, and ending with user technique (speaking directly into microphones and learning the etiquette of multi-site operation). However, audio problems are not insurmountable. Once your audio system has been correctly "tuned," you will find a dramatic increase in productivity and enjoyment.

Autobiographical Sketch

Craig Richardson received his Sc.B. in Electrical Engineering from Brown University in 1985 and his M.S.E.E. and Ph.D. in digital signal processing from the Georgia Institute of Technology in 1988 and 1992, respectively. In 1992 he joined the staff of ASPI Digital as the Director of Algorithm Development and in 1996 became President and CEO. He has led numerous real-time speech, audio, and video design and development projects and most recently led the design team for an MPEG-2 Layer III application for remote radio broadcasts. In addition to having written numerous articles and chapters in books, he is the co-author of the textbook (with Thomas P. Barnwell, III and Kambiz Nayebi) Speech Coding: A Computer Laboratory Textbook, a title in the Georgia Tech Digital Signal Processing Laboratory Series published by John Wiley & Sons.

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